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Efficient Implementation of VoIP Over VPN w.r.t Packet Delay and Data Security

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ABSTRACT

This research is about to build a network with Voice over Internet protocol (VoIP) traffic in Virtual Private Network (VPN) with efficient speed of data transfer rate and minimum packet delay. There are number of protocols that can be used for security enhancement and to reduce packet delay but in this study VoIP is implemented to test traffic on VPN for packet delay and data transfer rate. VoIP is a swiftly rising term for voice conversation that organizes VoIP client equipment's such as desktop IP phones, mobile VoIP-allowed handheld devices, control all functionality about voice communication. VPN still faces many issue regarding security and packet delay. Secured VPN may increase the quantity of packet loss. This proposed work provides minimum delay of packets and also increases the security performance with efficient codec and timeout in VoIP traffic over VPN. VoIP codec G.726, ilBC are also tested in this research. As a result both are best in Audio and video calls but depend on network infrastructure. Moreover, many tools such as (GNS3, OPNET, Wire shark), algorithms and methods are applied to checks the effect of VoIP traffic over VPN with respect to packet size, packet loss, timeout, time average to transfer file (download and upload), quality of services and best selection of code for VoIP traffic . Above cited codecs are compare the performance of software and hardware environment regarding VoIP over VPNs. This work increased the performance of data rate and minimized packet delay on VoIP trafficin VPN. Wire shark tool is used to analysis the data regarding packet on VoIP over VPN.

Keywords: *VPN, VOIP, GNS3, RTD, ilBC*

2. INTRODUCTIONS

VoIP is the new approach, group of transmission protocols under the kind shadow of Internet Protocol. VoIP still faces some challenges; the big challenges are data delay and packets loss as compared other internet application such as telephonic communication, email services and web. Moreover, another core issue is data security. Basic monitoring tools are used to measure the quality of VoIP traffic smoothly.(Ahmed, Yasuo, & Masanori, 2003) Some addressing schemes are improved the work of internet such as IPv4 over IPv6.

In this research, we will apply efficient VoIP codec schemes, architecture and topology plan on VoIP traffic in VPN to achieve a good performance result with respect to packet delay and give secured environment regarding data and communication. (Agrawal et al., 2006)This work is divided into different sections.

Section 2 gives the background work of VoIP over VPN and its performance factors. Section 3 is based on which kinds of materials are required and methods to achieve above abstract.(Anjum & Perros, 2015)In section 4 we will analyze the network using GNS3 and OPNET tools and also express the topology design plan. In last section is 5 in which give discuss the conclude work of this paper and future work.

2.1 Backgrounds

Secure and reliable platform is necessary for all networks which are based on VoIP. VoIP is a technology that is very popular now a day.(Bai & Ito, 2006) Virtual Private Networks is kind of private network that gives a well secure and reliable connection between nodes to nodes. VPN are best for secured communication IP networks and also supports the real time traffic. The major issues on VoIP are:

- Less secure
- Quality of services in voice transmission
- Packet loss
- Delay of packets
- Vulnerabilities

These are the all issues which may occur in communication between two nodes. Above mentioned major problems will also occur when VoIP deploy on large scale environment. Therefore implementation of VPN is needy in VoIP to overcome these hurdles during communication and also provide the efficient solution of packet loss, jitter and security gaps.

3. MATERIAL AND METHODS

Following tools are required to achieve the purpose of this research.

- Simulator (GNS3 or OPNET)
- Virtual Machine workstation
- Wire shark (Data Analyzer)
- Cisco Communication Manager
- Router’s IOS
- VPN Connection
- VoIP over VPN Connection

3.1 methodologies

Figure 1 show the topology of Virtual Private Network, which is based on three cisco-7200 routers one is represent as Internet Services Provider (ISP) and other are represent two different sides location. Figure 2 and show the topology of VoIP and figure 3 configure the VoIP over VPN using open source simulator GNS3. In this phase, analyze the performance factors of VPN, VoIP and VoIP over VPN with respect to packet loss, data delay, codec selection and secured communication. (Crawley, Sandick, Nair, & Rajagopalan, 1998)

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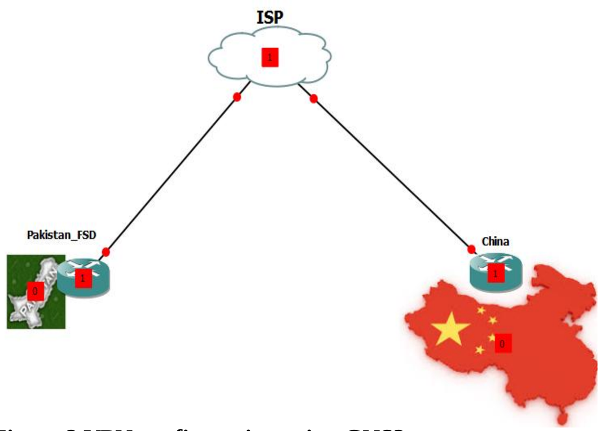

Figure 2 VPN configuration using GNS3

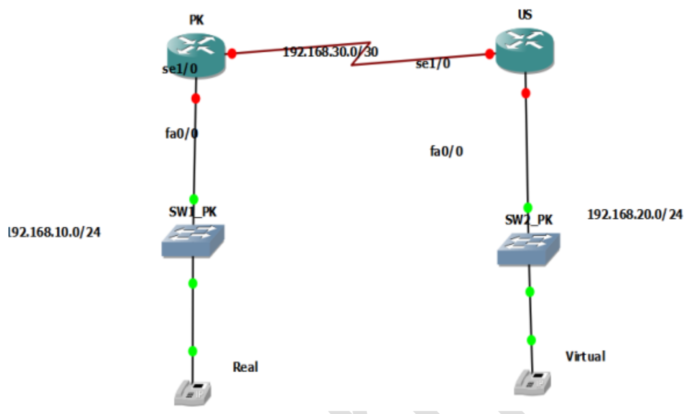

Figure 1 VoIP configuration using GN3

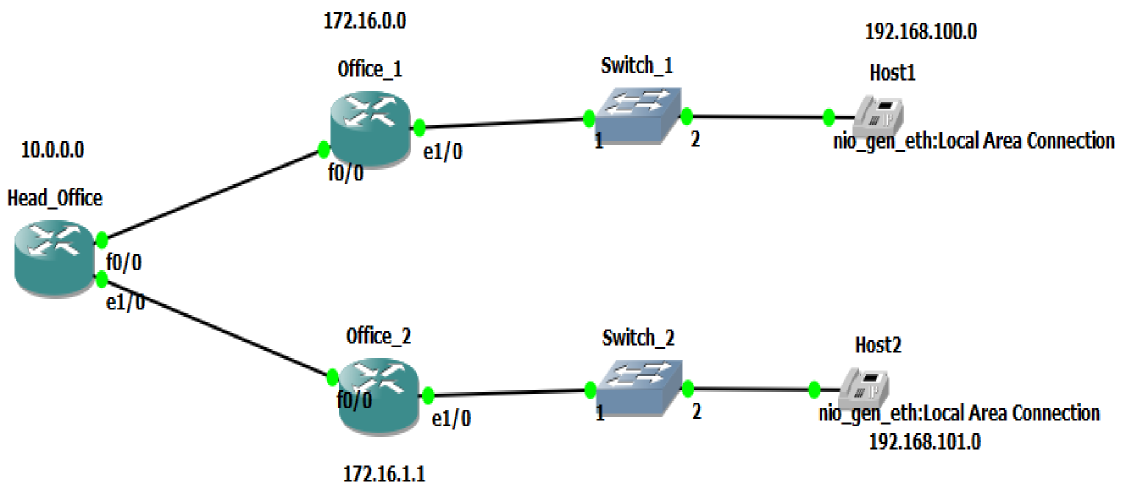

Figure 3 VoIP over VPN configuration

4. RESULTS AND DISCUSSION

Many tests are conduct to measure the performance of average time of packet size, RTD in packet delay, data transfer rate, packet loss, QoS in VoIP over VPN. (Thakur & Kumar, 2011)Moreover analyze the performance of CPU utilization, Memory and delay in performance.

4.1 Average time of packet length in voipover VPN

This result is produce by analyzer tool Wire shark. It takes data automatically from GNS3 simulator in which implement the VoIP over VPN. This result is proving by mathematically for user information.

Total Packet length is: 678 bytes and rate (ms) 0.002049

Table 1 Calculate Packet lenght size

Sub-Length of packets	Total numbers of items received	Arrival Time/ Rate(ms)	Average Time	Avg time/item	Best Avg
0-19	0	0.0000	0.000%	0.000%	
20-39	0	0.0000	0.000%	0.000%	
40-79	148	0.00001447	21.83%	0.295	
80-159	528	0.001595	77.88%	0.295	
160-319	2	0.000006	0.29%	0.29	0.29

Sub-length: Is the item received (specified packet type in calls from source to destination)
Number of items received: is the number of this item received.
Average Time: is the amount of packets of this type that make up the total count.
Arrival time/Rate (ms): the rate the packets arrive in milliseconds?

4.1.1 Calculation formula for arrival time/rate (ms) and average time.

Arrival time/rate (ms) = Total Packet Length/rate (ms)/1000 answer of this formula to apply *Total received items/rate(ms)1000.*

Average Time = Number of items Received/Total Packet Length*100

4.2 RTD- packet delay in voipover VPN

In this result target a specific IP-address to determine the packet delay and reliability of connection between networks.(Shi & Turner, 2002) In following result found that when timeout volume is increase the efficiency of round-trip is increase and there is less chance to loss the packet as well as delay in packets capturing.

Table 2 Calculate RTD w.r.t packet loss and timeout

Target IP	Rep. Count	Datagram Size	Timeout in seconds	Loss Packet	Round-Trip(ms)		
					Min	Max	Avg
10.0.0.0	5	100	3	0	180	504	342
10.0.0.0	8	50	2	1	136	320	228
172.16.0.0	5	100	3	3	244	584	427
172.16.0.0	5	50	30	1	367	840	576
172.16.1.1	5	100	3	3	244	584	427
172.16.1.1	5	50	30	1	367	840	576

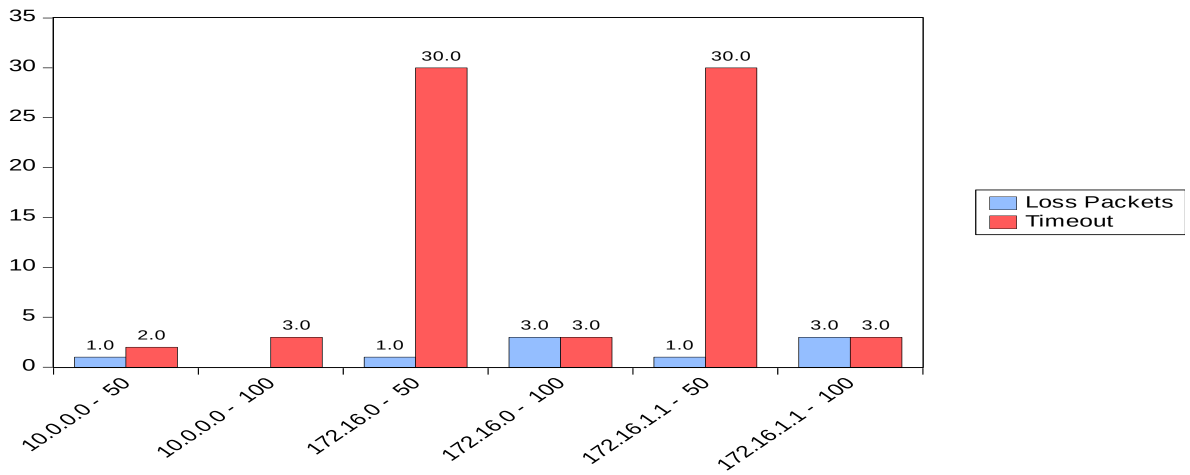

Figure 4 Loss packets vs Timeout

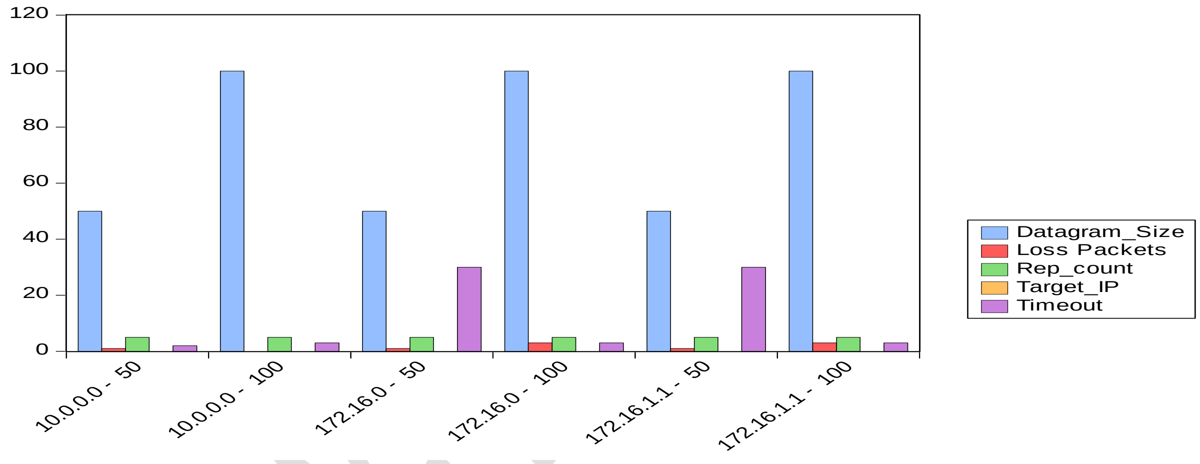

Figure 5Result of packet delay against IP

4.3 Average time of Data Transfer in voipover VPN

Position:	**Data transfer Details (bytes in seconds)**
Status:	Data is transfer successfully 211,057,507 bytes in 211
Status:	Data is transfer successfully 211,057,507 bytes in 202
Status:	Data is transfer successfully 211,057,507 bytes in 195
Status:	Data is transfer successfully 211,057,507 bytes in 270
Status:	Data is transfer successfully 211,057,507 bytes in 250
Status:	Data is transfer successfully 211,057,507 bytes in 207

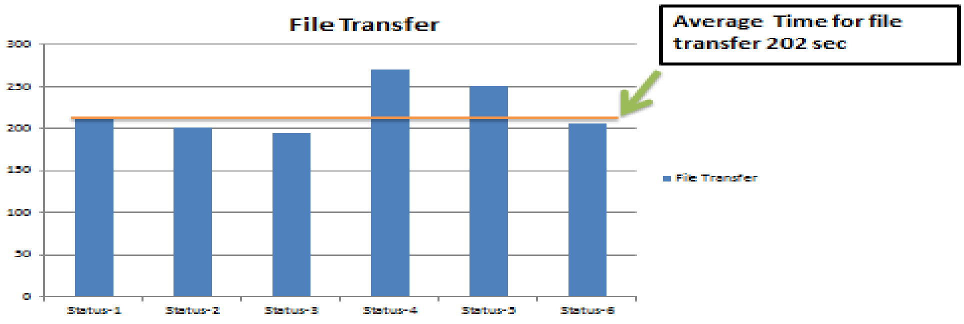

Figure 6 Data Transfer rate

The above figure is showing the status of uplink file and downlink file in VoIP over VPN, least and most extreme document exchange time is 195 seconds (3.250 minutes) and 270 seconds (4.5 minutes). Normal time for document move in this investigation is 202 seconds (3.36 minutes) and standard deviation is 27.54.(Lai, Gao, Ma, Nong, & Wang, 2011)

4.4 Codec Selection w.r.t packet delay and system performance

Many codecs are used in network and operating system. Some famous codecs are G.711, G.726, G 7.29, and Internet Low Bitrate Codec (ilBC) used in VoIP to overcome the packet loss and delay in network. These codecs are test in Wireshark tools according to result G.7.26 and ilBC are best for VoIP to give good performance as compare to other codecs.(Chen, Yang, & Xu, 2004)

Table 3 selection Codec of system for VoIP over VPN

Codec	G711	G726	ilBC
CPU Utilization	High	High	High
Memory Performance	Medium	Medium	Low
Delay	High	High	Low
Jitter	High	High	High

5. CONCLUSIONS AND FUTURE WORK

This study tells the packet delay issue in VoIP environment and the performance of VoIP over VPN network. VPN used as a backbone in VoIP environment. This research is also investigates the packet delay, data security on VoIP over VPN and defines the methods to overcome these issues. One thing should keep in this methodology that select best VoIP codec schemes regarding software and hardware.(Alshammari & Zincir-Heywood, 2015)In VPN many issues will be face regarding latency but this is also overcome to increase the timeout in network. There for it is strongly recommended to use VPN for VoIP traffic. Efficient and effective techniques are very important to implement the VPN services on VoIP network in order to achieve good services and security issues. In future work, implantation of this research will be deployed on mesh network.

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